REMARKS

The specification has been amended to correct errors of a typographical and grammatical nature. Due to the number of corrections thereto, applicants submit herewith a Substitute Specification, along with a marked-up copy of the original specification for the Examiner's convenience. The substitute specification includes the changes as shown in the marked-up copy and includes no new matter. Therefore, entry of the Substitute Specification is respectfully requested.

The claims and abstract have also been amended to more clearly describe the features of the present invention.

Entry of the preliminary amendments and examination of the application is respectfully requested.

To the extent necessary, applicant's petition for an extension of time under 37 CFR 1.136. Please charge any shortage in the fees due in connection with the filing of this paper, including extension of time fees, to Deposit Account No. 01-2135 (Case: 501.42784X00) and please credit any excess fees to such deposit account.

Respectfully submitted,

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TITLE OF THE INVENTION

MESSAGE CONVERSION SERVER AND IP TELEPHONE

BACKGROUND OF THE INVENTION

Field of the Invention

The present invention relates to a calling method for used when a caller makes a call to a callee in a one-to-one communication mode on the Internet.

Non-patent documents listed below are referred to in the following description in a following paragraph.:

[Non-patent document 1]

IETF RFC3261

[Non-patent document 2]

IETF RFC2327

[Non-patent document 3]

IETF RFC1889

In recent years, there has been a tendency to integrate exchange-based telephone communications networks into IP networks has been developed as part of with a rapid advance in IP network technology. Telephone communications carriers have a plan to transfer data under voice (hereafter, simply referred to as DUV) on their own IP networks through Voice over IP (hereafter, simply referred to as VoIP). VoIP consists of two protocols, one protocol controlling call signaling and sessions and the other protocol controlling DUV transfer. The

Internet Engineering Task Force (hereafter, simply referred to as IETF) made a specifications of Session Initiation Protocol (hereafter, simply referred to as SIP), IETF RFC3261, defining the method for call signaling and sessions. For example, Session Description Protocol (SDP), IETF RFC2327, is applicable to the description of a session including an agreement on a codec and transfer rate to be internally used in SIP.

Although no specific DUV protocol is defined in SIP, Real-time Transport Protocol (RTP), RFC1889, is commonly used.

According to IETF RFC3261, SIP is such a protocol that a SIP message consisting of a SIP Start Line, a SIP message header, and a SIP message body is sent/received between two calling parties via a SIP server to make an agreement about the call signaling mode on the callee side, the voice, image protocol and bit rate to be used during conversation. Generally, the SIP Start Line describes the behavior of the message originator, the SIP message header describes the telephone number of a callee, the SIP server passed, and Call-ID (call origination administration number), and the SIP message body describes the proposed voice, image protocol, and bit rate to be used during conversation.

Now, a procedure is described in brief ranging from the start to end of a two-party conversation through SIP

described in IETF RFC3261, and the problems which arise with use of the procedure will be ~~are~~ explained.

Fig. 1 is a network diagram illustrating a ~~showing~~ two-party conversation carried out through use of SIP.

Fig. 2 is a sequence diagram illustrating the ~~explaining a~~ flow of a two-party conversation through use of SIP.

In Fig. 1, UserA, who belongs to a domain 3-1 and has an IP Telephone 2-1, makes a call to UserB, who belongs to a domain 3-3 and has an IP Telephone 2-2, via SIP servers 1-1 to 1-3.

First, UserA sends a Start Line INVITE and a SIP message for UserB to the SIP server A1-1 to establish a call with UserB (11). The SIP server A1-1, when receiving the INVITE message, adds a VIA header to its message header and transfers the SIP message to the SIP server B1-2. At that time, it also sends the SIP message containing a Start Line 100Trying to the IP Telephone 2-1, which is the callee (destination) of the message (12). The SIP servers B1-2 and C1-3, when receiving the SIP message, take the same actions and transfer the message to the UserB's IP Telephone 2-2.

The IP Telephone 2-2, when receiving the SIP message, sends a Start Line 180Ringing and a SIP message for UserA to the SIP server C1-3 (13) to sound a ringing tone on the UserB side. The SIP message containing a Start Line

180Ringing terminates at the IP Telephone 2-1 via the SIP server.

The IP Telephone 2-2, when UserB picks up ~~a~~ the telephone receiver, sends a Start Line 200OK and a SIP message for UserA (14), which in turn, terminates at the IP Telephone 2-1 via the SIP server. The IP Telephone 2-1 sends back an acknowledge (ACK) signal in response to the message (15), and, when the ACK is received, a voice packet passes through a main signal path, enabling the two parties to initiate a conversation between them (16).

At the end of the conversation, UserA's IP Telephone 2-1 sends a Start Line BYE and a SIP message for UserB (17), which in turn, ~~determines~~ terminates at the IP Telephone 2-2 via the SIP server. In response to the message, the IP Telephone 2-2 sends back the ACK to the IP Telephone 2-1 via the SIP server (18). When the ACK is received, the conversation ends.

SIP is a protocol for sending and receiving SIP messages between a caller and a callee. A UserID and its DomainID are described in the headers, "From" specifying the caller contained in the message header, "To" specifying the callee, and "Via" specifying the SIP servers to be passed (proxy mode), because ~~these~~ such information can be delivered as ~~they are~~ it is when the callee sends it back. To establish a session with

the callee through SIP, the caller describes the IP address of his/her own terminal or an assigned DomainID in the headers.

With regard to VoIP, a protocol which informs the callee of no UserID (e.g. telephone number or UserID) of the caller, such a mode has been proposed in IETF RFC3261 that whereby the caller terminal creates a random UserID, registers the random UserID and the IP address of the terminal in the SIP server, and originates a call with the random UserID designated as the caller. Through this protocol, the whole procedure for making a Caller Anonymous Call is performed on the caller side.

If the IETF-supported communication mode, in which the caller makes a Caller Anonymous Call with his/her UserID concealed, is used in the SIP system, only the random UserID is registered in the SIP server. For this reason, the caller would make a Caller Anonymous Call not only to the callee, but also to the SIP server at the same time. The SIP server is difficult to control, and it is difficult to manage calls because it cannot guess the real UserID from the UserID registered in it. In the IP telephone services provided by communications carriers and others, user management is required, for example, caller identification, service permission/denial determination, and talk time management.

If a malicious third party intercepts a SIP message, he/she can know identify the caller, the caller's SIP-URL, and the

assigned DomainID as described in the SIP message, causing mischief, ~~damages~~ such as nuisance calls.

In addition, if the malicious third party knows the IP address of the IP Telephone, he/she can transfer a vast amount of packets to the IP Telephone after the end of conversation, making an attack, for example, DOS (Denial of Service), thereby disturbing processing on the equipment at a high possibility.

SUMMARY OF THE INVENTION

An object of the present invention is to provide a method of effecting a "Non caller informed call", which enables the SIP servers to manage caller information (UserID identifying the UserID of the caller sending the SIP message and his/her DomainID), while concealing the information from the callee and ~~the~~ a malicious third party.

Another object of the present invention is to make it difficult for ~~the~~ a third party to identify the IP address assigned to the IP Telephone, thereby to minimize any ~~damage~~ problems he/she may cause when he/she knows the IP address.

According to one aspect of the present invention, a packet forwarding device~~,~~ transmits a message sent by a caller to a specified callee, wherein the device has a processing part for providing at least either ~~the~~ a function for converting or ~~the~~ a function for erasing at least part of

the message sent by the caller upon his/her request and a control part for determining whether at least the part of the message should be converted or erased, and it converts or erases at least the part of the message based on the determination in the control part. This mechanism enables information identifying the caller to be concealed from the callee.

At least the part of the message that is converted or erased, as described above, may be one of or any combination of:

(1) the a part identifying the user on a calling side in the SIP message header on an IP packet payload containing the message[,];

(2) a the part identifying the caller's domain in the SIP message header on the IP packet payload containing the message[,];

(3) a the part of a Via tag in the SIP message header on the IP packet payload containing the message[,];

(4) the a part indicating the Call-ID domain in the SIP message header on the IP packet payload containing the message[,]; and

(5) the a part identifying a UserID in the SIP message body on the IP packed payload containing the message.

It may be possible that for the control part to analyzes the content of the message, when receiving it[,]; and, if any given character string or header is detected, the processing part operates to converts or erase part of the message. Any given character

string is a series of numeric characters contained in the position of the first numeric string, for example "184". Any given header is the SIP message header, and, when its extended header is detected, the above-mentioned processing part may convert or erase part of the message.

Additionally, it is preferable that to provide a table, in which the correspondence between the contents of part of the message before and after conversion is contained, is prepared. According to one aspect of the present invention, to conceal information on SIP-URL and the assigned domain of the caller, the SIP server that is installed at a relay point between the caller and the callee converts the SIP message. The SIP server with the message conversion function is characterized in that it provides the a method for converting or erasing part of the message originated by the caller, the a processing part for determining whether it should be converted or erased or not, the a processing part for determining information to be used in conversion, and the a table containing the rule of conversion.

According to the an aspect of the present invention, in order to make it difficult for the a third party to identify know the IP address of a caller's IP Telephone, the IP Telephone creates or obtains a temporary IP address to be used only once when the SIP message is sent and discards it as soon as the conversation ends up. The IP Telephone is characterized in

that it provides ~~the~~ a method for creating or obtaining the IP address in conjunction with the origination of the SIP message and for discarding it as soon as the conversation ends, ~~the~~ a processing part for registering the temporary IP address in the SIP server, ~~the~~ a processing part for canceling its registration from the SIP server, and ~~the~~ a processing part for creating a random interface ID.

It is possible that, optionally, the step ~~for~~ of converting the SIP message using the SIP server and/or the step ~~for~~ of obtaining the temporary IP address using the IP Telephone may be used. According to another aspect of the present invention, the SIP telephone communication method involves a step ~~for~~ of receiving ~~the~~ an SIP message, a step ~~for~~ of checking the SIP message for any request for Anonymous Call, a step ~~for~~ of executing at least one of two operations, including modification and deletion on at least part of the SIP message if any request is detected, and a step ~~for~~ of sending the SIP message that has been processed as described above. It is preferable that, if the request for Anonymous Call is detected, a modification is made on at least part of the SIP, and a table containing the correspondence between the contents before and after modification is created.

According to ~~a~~ a second ~~another~~ aspect of the present invention, the IP telephone communication method involves a process for modifying the caller address to ~~its~~ a temporary address at the

initiation of conversation and ~~the~~ a process for discarding the ~~above~~ temporary address at the end of conversation.

According to a third ~~another~~ aspect of the present invention, the method for converting or erasing part of ~~the~~ a message sent by the caller upon his/her request is characterized by a step ~~for~~ of determining whether part of the message should be converted or erased, a step ~~for~~ of determining information to be used in conversion, if it is determined to be modified, and a rule applicable to the conversion. In addition, a telephone set is characterized in that it involves a method which has step ~~for~~ of modifying the address of a caller in conjunction with origination of the message every time he/she makes a call to prevent ~~it~~ the address from being disclosed and a step ~~for~~ of disposing the address indicating the different UserID that is used at the time of conversation as soon as the conversation ends~~;~~, and the telephone set further provides ~~the~~ a method for assigning the addresses indicating different UserIDs at the time of call origination and call receiving and ~~the~~ a method for sending the user's call. The scope of the present invention includes the methods, devices, and systems described above.

As explained above, the invention achieves ~~the~~ a function that is compatible with ~~the~~ an exchange-based Anonymous Call function in ~~the~~ an IP Telephone.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a schematic diagram ~~view~~ showing an example of ~~the~~ a SIP network ~~created for explaining the object~~ of the type to which the present invention is applicable;

Fig.2 is a sequence diagram showing an example of ~~the~~ a procedure for making a call in the SIP network of Fig 1 ~~created for explaining the object of the present invention~~;

Fig. 3 is a schematic diagram ~~drawing~~ showing an example of the SIP network of the present invention;

Fig. 4 is a flow chart showing the operational principle of ~~the~~ an SIP server ~~with~~ which has the message exchange function of the present invention;

Fig. 5 is a block diagram of ~~the~~ an SIP server ~~with~~ which has the message exchange function of the present invention;

Fig. 6 is a block diagram of ~~the~~ an SIP server ~~with~~ which has the message exchange function of the present invention;

Fig. 7 is a diagram ~~view~~ showing the format of ~~the~~ an IP packet containing ~~the~~ an SIP message;

Fig. 8 is a schematic diagram ~~view~~ showing an example of ~~the~~ a network using an ~~the~~ SIP server ~~with~~ which has the message conversion function of the present invention.

Fig. 9 is a sequence diagram showing ~~the~~ a procedure for making a call in the example of the network using ~~the~~ an SIP server ~~with~~ which has the message conversion function of the present invention;

Fig. 10 is a diagram ~~view~~ showing the contents of a SIP message header that is unconverted and that is converted by ~~the~~ a SIP server ~~with~~ which has the message conversion function of the present invention;

Fig. 11 is a diagram ~~view~~ of the content of a SIP message body (SDP) that has been unconverted by the SIP server with the message conversion function of the present invention;

Fig. 12 is a diagram ~~view~~ of the content of a SIP message body (SDP) that has been converted by the SIP server with the message conversion function of the present invention;

Fig. 13 is a diagram ~~view~~ showing the content of the conversion table stored on the SIP server with the message conversion function of the present invention;

Fig. 14 is a diagram ~~another view~~ showing another example of the content of the conversion table stored on the SIP server with the message conversion function of the present invention;

Fig. 15 is a schematic diagram ~~view~~ of ~~the~~ a network using ~~the~~ a SIP server ~~with~~ which has the message conversion function of the present invention;

Fig. 16 is a sequence diagram showing the procedure for making a call in the network using the SIP server with the message conversion function of the present invention;

Fig. 17 is a diagram ~~view~~ showing the contents of the SIP message header that is unconverted and that is converted by ~~the~~ a SIP server ~~with~~ which has the message conversion function of the present invention;

Fig. 18 is a ~~view~~ diagram showing the content of a conversion table stored on ~~the~~ a SIP server ~~with~~ which has the message conversion function of the present invention;

Fig. 19 is a diagram ~~another view~~ showing another example of the content of the conversion table stored on ~~the~~ a SIP server ~~with~~ which has the message conversion table of the present invention;

Fig. 20 is a schematic diagram ~~view~~ showing ~~the~~ a network using ~~the~~ a SIP server ~~with~~ which has the message conversion function of the present invention;

Fig. 21 is a sequence diagram showing the procedure for making a call in the network using the SIP server ~~with~~ which has the message conversion function of the present invention;

Fig. 22 is a diagram ~~view~~ showing the contents of the SIP message that is unconverted and that is converted by the SIP server ~~with~~ which has the message conversion function of the present invention;

Fig. 23 is a diagram ~~view~~ showing the content of the conversion table stored on ~~the~~ a SIP server ~~with~~ which has the message conversion function of the present invention;

Fig. 24 is a diagram ~~another view~~ showing another example of the content of the conversion table stored on the SIP server ~~with~~ which has the message conversion function of the present invention;

Fig. 25 is a diagram ~~view~~ showing ~~the~~ a network using ~~the~~ a SIP server ~~with~~ which has the message conversion function of the present invention;

Fig. 26 is a sequence diagram drawing showing the procedure for making a call in the network using a the SIP server with the message conversion function of the present invention;

Fig. 27 is a diagram view showing the contents of the SIP message header that is unconverted and that is converted by the SIP server with the message conversion function of the present invention;

Fig. 28 is a flow chart showing the principle of the operation of the IP Telephone of the present invention, ranging from the step for of obtaining the temporary address to the step for of discarding it;

Fig. 29 is a block diagram showing the function of the IP Telephone of the present invention;

Fig. 30 is a systematic diagram of IPv6 addresses; and

Fig. 31 is a diagram view showing the content of the a SIP message header when a Caller Anonymous Call is made.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Embodiment 1

Fig. 3 is a view show~~ing~~s a first embodiment of an IP telephone line network using a SIP server having the with a message exchange function of the present invention. Fig. 4 is a flow chart illustrating the explaining an operational procedure of the SIP server 12-1 of Fig 3 with the message exchange function.

Fig. 5 is a functional block diagram explaining the function of the SIP server with the message exchange function 12-1.

Fig. 6 is a block diagram view showing the hardware configuration of the SIP server with the message exchange function 12-1.

Fig. 7 is a diagram view showing an IP packet 60 containing a SIP message. They are by The IP packet is represented by 60, an IPv4/v6 header by 61, a TCP/UDP header by 62, and a payload 63, which includes a SIP Start Line by 64, a SIP message header by 65, and a SIP message body (SDP) by 66.

Now, referring to Fig. 4 and Fig. 5, the operational principle of the SIP server 12-1 with the, which has a message exchange function, will be is described below.

First of all, The IP packet indicated in Fig. 7 is input via IF 51. Second, receipt receiving of the SIP massage 22 is performed at a SIP message sending/receiving part 45, and a Start Line check 23, a Header check 24, and a body check 25 are performed at a by the SIP message checking part 44. If any error is detected in the SIP message, the process ends with the issuing of an error response notification 33 issued.

If no error is detected, then a message conversion check 26 is performed at during a message conversion request check 26 at a message conversion/processing part 47.

If no conversion request is detected, a Via header in is described in the SIP message header 64 at the message conversion exchange/processing part 47, and then the SIP message is sent

via IF51. If ~~the~~ a conversion request is issued with a flag 561 and 571 (see Fig.31) indicating that ~~the~~ a "No caller informed call" is to be originated, the step ~~for converting~~ ~~the~~ message header conversion 27 and the step ~~for~~ of converting the message body 28 are performed at the message conversion/processing part 47. These steps conceal information on the caller from ~~its~~ the callee (destination). After the conversion, a step of ~~for~~ extracting conversion information 29 is performed at a conversion information extraction/transfer part 46 to pick up header body information necessary for creating a conversion table. Then, a step ~~for~~ of creating the conversion information table 30 is performed at a conversion entry creation/modification part 49, writing into the conversion information table 31 is performed at a conversion entry I/O part 48, and the converted entry is registered at a conversion entry registration part 50. The converted SIP message undergoes message transfer 32 at a SIP message sending/receiving part 45 via IF51.

According to the embodiment of the present invention, the function of the conversion entry creation/processing part 49 in the SIP message processing part 41 and the conversion table processing part 42 shown in Fig. 5 ~~is~~ are executed on the CPU72 shown in Fig. 6. The function of the conversion entry I/O part 48 shown in Fig. 5 is executed at

a conversion table fetching part 75 shown in Fig. 6. The function of the conversion table registration part 43 shown in Fig. 5 is executed at a conversion table storage part 74 shown in Fig. 6.

Embodiment 2

Now, a more detailed embodiment is described below.

Fig. 8 is a schematic view of a network configuration showing the second embodiment of the SIP server of the present invention.

Fig. 9 is a sequence diagram explaining the communication procedure in the embodiment of the network using the SIP server with the message conversion function of the present invention.

In the second embodiment, UserA makes a Caller Anonymous Call to UserB. The step for processing the Non caller informed call is performed on the SIP server A12-2 with the message exchange function. The step for converting the SIP message is performed on the SIP server A (12-4) with a header conversion function used by the caller in sending the SIP message at the steps 111 and 112 in the sequence ranging from the start to the end of conversation as shown in Fig. 9. Note that the SIP server with the conversion function behaves as shown in the first embodiment. Fig. 10,

Fig. 11, and Fig. 12 show the content of the converted SIP message in the second embodiment.

Fig. 10 is a diagram view showing the header part of the SIP message (65 in Fig. 7). In the upper part, the unconverted header is shown and in the lower part, the converted header is shown. In the second embodiment, a UserID 142 of the From tag and a part identifying the user of SIP-URL 143 in the unconverted header 141 are converted into, for example, character strings 147 and 148 such as Anonymous, from which the UserID of the caller cannot be guessed.

Fig. 11 is a diagram view showing a body part of the SIP message (Unconverted). The body part of the SIP message is represented by 66 in Fig. 7.

Fig. 12 is a diagram view showing the body part of the SIP message (converted). In this figure, the part identifying a UserID 152 of the message body (SDP) 151 of Fig 11 is converted into Anonymous 156. The part 153 of Fig 11, identifying the user's address to be used during conversation in an "o","c" tag is converted into the IP address if described in FQDN (Fully Qualified DomainID).

As described above, the present invention enables information on the caller to be concealed by converting information on the caller, based on which the callee can identify the UserID of the caller including the UserID 142 of the From tag, the part identifying the SIP-URL user 143,

and the part 152 identifying the UserID of the message body (SDP).

Fig. 13 and Fig. 14 are diagrams views showing the conversion tables stored on the SIP server 12 with the conversion function in the first embodiment of the present invention. These conversion tables include the table 170, which associates all the UserIDs converted into the same Call-ID 172 with their real UserIDs 171, the table 180, which associates anonymous UserIDs 181 previously stored on the SIP server as "Anonymous1-100" with their real UserIDs 182, and others. Reference to these tables associating anonymous UserIDs 181 with Call-ID 172 makes it possible to effect successful routing of the SIP messages.

Embodiment 3

Fig. 15 is a schematic diagram view showing the network configuration in a the third embodiment of a the SIP server which has with the message conversion function of the present invention. Unlike the embodiment shown in Fig. 8, in this embodiment, anonymous processing is executed on the SIP server ~~8~~ c. Fig. 16 is a sequence diagram showing the procedure for processing conversation on the network of Fig 15 using the SIP with the message exchange function of the present invention.

In the this third embodiment of the present invention, UserA makes a Callee Anonymous Call to UserB. The step for of

converting the SIP message is performed on the SIP server C (12-3) with a header conversion function, which sends the SIP message at the steps 211 and 212 in the sequence ranging from the start to the end of conversation, as shown in Fig. 16. Note that the SIP server with the conversion function behaves as described ~~shown~~ in connection with the first embodiment.

Fig. 17 is a diagram ~~view~~ showing the content of the SIP message converted on the SIP server with the message conversion function. Both of the unconverted and converted message headers are shown (65 in Fig. 7).

In this ~~the~~ third embodiment of the present invention, the Via tag 222 is erased and only the Via tag 232 of the ~~an own~~ server itself is described in the unconverted message header 221. The UserID 223 in the From tag is converted into the tag 233, from which no UserID of the callee can be guessed, such as Anonymous, and the part 224 identifying SIP-URL is converted into part 234 ~~225~~, from which no UserID and its domain can be guessed. In addition, the part 225 identifying the Call-ID's domain is converted into part 235. The rule of message body (SDP) conversion is the same as that of the first embodiment of the present invention.

As described above, in the embodiment shown in Fig. 17, the Via tag indicating the relay point for the message can be deleted to prevent the call source from being guessed. Alternately, the part 225 identifying the Call-ID's domain

can be converted into temporary DomainID 235, from which no domain can be guessed.

Fig. 18 and Fig. 19 are views showing the conversion tables stored on the SIP server 12-3 with the conversion function in the second embodiment of the present invention. These conversion tables include the table 271, which associates all the UserIDs converted into the same Call-ID 273 with their real UserIDs 272, the table 281, which associates anonymous UserIDs 282 previously stored on the SIP server as "Anonymous1-100" with their real UserIDs 283, and others. The use of these tables in routing SIP messages conceals information from the callee, ensuring information security.

Embodiment 4

Fig. 20 is a view showing the network configuration of the SIP server in the fourth embodiment of the present invention.

Fig. 21 is a sequence diagram showing the procedure for processing conversation in the network using the SIP server with the message conversion function.

In the fourth embodiment of the present invention, UserA makes a Caller Anonymous Call. The step for converting the SIP message is performed on the SIP server A (12-4) with the header conversion function used by the caller in sending

the SIP message at the steps 311 and 314, and on the SIP server C (12-5), with the header conversion function at the steps 312 and 313, which sends the SIP message to the callee in the sequence ranging from the start to the end of conversation, as shown in Fig. 21. Note that the SIP server with the conversion function behaves as described shown in connection with the first embodiment.

Fig. 22 is a diagram view showing the content of the SIP message converted on the SIP server with the message conversion function. In this the fourth embodiment of the present invention, first, the UserIDs 322 and 323 of the From tag in the unconverted header 321 are is converted into character strings 326 and 327, from which no UserID of the callee can be guessed, such as Anonymous, and the message body (SDP) is converted in accordance with the same rule as that of the first embodiment of the present invention. Second, the Via tag 332 in the unconverted header 331 is erased on the SIP server 12-5 and only the Via tag 336 of the an own server itself is described. The part 333 identifying SIP-URL in the From tag is converted into part 337, from which no DomainID can be guessed. The part 334 identifying the Call-ID's domain is converted into part 338.

Fig. 23 is a diagram view showing the conversion table stored on the SIP server 12-4 with the conversion function.

Fig. 24 is a view showing the conversion table stored on the SIP server 12-5 with the conversion function.

The conversion tables stored on the SIP server 12-4 include the table 341, which associates all the UserIDs converted into the same Call-ID 343 with their real UserIDs 342, the table 351, which associates anonymous UserIDs 352 previously stored on the SIP server as "Anonymous1-100" with their real UserIDs 353, and others. The conversion tables stored on the SIP server 12-5 include the table 361, which associates Call-ID 362 with DomainIDs 364, and others.

Embodiment 5

Fig. 25 is a schematic diagram ~~view~~ showing the network structure in ~~the~~ a fifth embodiment of the SIP server of the prevent invention. Fig. 26 is a sequence diagram showing the procedure for processing conversation in the network of Fig 25 ~~using the SIP server with the message conversion function of the present invention~~.

In the fifth embodiment of the present invention, UserA make a Caller Anonymous Call to UserB. The step ~~for~~ of converting the SIP message is performed on the SIP server ~~A~~ C (12-6) with a header conversion function belonging to the top level domain for each communications carrier at the

steps 411 and 412 in the sequence ranging from the start to the end of conversation as shown in Fig. 26.

Fig. 27 is a diagram view showing the content of the SIP message converted on the SIP server with the message conversion function in the fifth embodiment of the present invention. In the fifth embodiment of the present invention, the Via tag 421 is erased and only the Via tag 426 of the an own server itself, the communication carrier, is described in the unconverted message header 420. The UserID 422 and the part 423 identifying user's SIP-URL in the From tag are converted into character strings 427 and 428, from which no UserID of the callee can be guessed, such as Anonymous. In addition, the part 424 identifying the caller's domain in the Call-ID tag is converted into the top level domain 429 of the communication carrier. The rule of message body (SDP) conversion is the same as that of the first embodiment of the present invention. The conversion tables stored on the SIP server 12-6 with the conversion function in the fourth embodiment of the present invention are shown in Fig. 18 and Fig. 19 and the contents of these tables are the same in those of the second embodiment of the present invention.

Embodiment 6

Fig. 28 is s flow chart [explaining] illustrating the operational principle of the IP Telephone using the temporary IP address at conversation.

Fig. 29 is a block diagram illustrating [explaining] the function of the IP Telephone. The operational principle of the IP Telephone in the sixth embodiment of the present invention will be [is] described below.

Fig. 30 is a [systematic] diagram of IPv6 addresses.

Fig. 31 is a view showing the SIP message header used when a Caller Anonymous Call is made.

First, the procedure for initiating a call [is] will be described [below]. When the caller originates a call to another user, the IP Telephone 521 initiates the step [for] of sending the SIP message and executes the step 502 of [for] selecting the address acquisition method [502]. If a random address creation method is selected, then the step 503 of [for] sending Router Solicitation [503] is performed to obtain an IPv6 address prefix 551 from a router in the same subnet. When the router sends Router Advertisement in response to Router Solicitation, a step 505 of [for] receiving Router Advertisement [505] is performed to obtain the address prefix 551.

Second, a step [for] of creating an interface ID 506 is performed at the random interface ID creation part 528 to create an IPv6 interface ID552.

no ¶ ~~The~~ Examples of the address prefix and the interface ID are represented by 553 and 554 in Fig. 30.

Third, a step ~~for~~ 507 of creating an IP address ~~507~~ is performed at a temporary IP address processing part 529 using the address prefix 501 and the interface ID552.

If the option of acquisition from the DHCP server is selected at the step ~~for~~ 502 of selecting the address acquisition method ~~502~~, a request for address acquisition 504 is issued to any address distribution server, for example, the DHCP server, to execute a step ~~for~~ 508 of obtaining the temporary IP address 508.

Whenever an IP call is made through IPv4, the address should be obtained from an external server..

Fourth, the modified entry or new registration entry of user information is ~~crested~~ created at a user data processing part 532 using the temporary address and the UserID to execute a step ~~for~~ 509 of registering the user's account ~~509~~.

Fifth, a step ~~for~~ 510 of creating the SIP message ~~510~~ at a SIP message header creation part 531 and a step ~~for~~ 511 of creating the SIP message body ~~511~~ at a SIP message body creation part 530 are performed, respectively.

As shown in Fig. 31, if it is desired to making a call with the UserID of the caller concealed from the callee, a flag indicating the SIP server ~~that~~ through which a Caller Anonymous Call is to be made is described in the SIP message header 560,

for example, a numeric value 184 (561) attached to the position directly before the telephone number of callee in the case of making a Caller Anonymous Call at an exchange-based telephone system or extended header (571).

Then, the steps ~~for~~ of creating a SIP Start Line INVITE, indicating a request to the callee, and creating the SIP message at a SIP signaling generation part 525, and the steps ~~for~~ of creating the IP packet 60 and sending the DIP signal 512 at an IP packet processing part 526 are performed, respectively, to initiate conversation.

At the end of the conversation, a step ~~for~~ 514 of erasing the account registration entry ~~514~~ is performed at the user data processing part 532 ~~t~~ to erase the account from the SIP server, and a step ~~for~~ 515 of discarding the IP address ~~515~~ is performed at the temporary IP address processing part 529, respectively, to complete the process.

The procedure for receiving the SIP message is the same as that used for sending it, with ~~an~~ the exception that the step ~~for~~ of obtaining the temporary address and the step ~~for~~ of registering the account are performed when the IP Telephone is powered on, or, when the IP Telephone logs in the domain managed by the SIP server, the SIP message is received, conversation is carried out ~~made~~, the temporary IP address is discarded at the end of the conversation, and immediately thereafter ~~after then~~,

a new temporary IP address is obtained for account registration.

As ~~known~~ can be seen from the description set forth above, the IP Telephone has two temporary IP addresses, one for sending and one for receiving, while the IPv4 telephone set has either one of them, because two addresses cannot be set on one terminal at the same time.

The callee receiving the SIP signal from the SIP server with header conversion function according to the embodiment of the present invention described above can recognize that the caller ~~makes~~ is making a Caller Anonymous Call by checking the converted UserID indicating ~~an~~ anonymous in the SIP message.

If the DomainID in the SIP message has been converted or erased for concealing one party's information from the other, the callee receiving the SIP message cannot know the caller's domain. Thus, a ~~The~~ malicious third party, even when receiving the SIP message sent by the caller, ~~is~~ finds it difficult to guess the caller because the UserID is concealed.

The callers can be managed by any organization, for example, a communications carrier, because the SIP server contains the conversion tables associating ~~between~~ real UserIDs ~~and~~ with their other parameters.

With the IP Telephone using ~~the~~ temporary IP addresses according to the embodiment of the present invention, the

IP address is modified for each call, making it difficult ~~it~~ for ~~the~~ a malicious third party to guess the identity of the caller even when he/she intercepts the IP packet during conversation.

In addition, when the SIP message is sent through IPv6, it is difficult for the third party ~~is difficult~~ to guess the identity of the caller because many IP addresses are described in the same segment.